IN THE SPECIFICATION:

Page 1, second paragraph, amend:

--However, one limiting factor in speech recognition is that the speech level, and thus the signal/noise ratio, decreases with an increasing distance between the sound source and the microphone. In environments with undesired interfering noise sources, such as cockpits in airplanes, motor vehicles, conference rooms, lecture halls and surgery rooms, it is therefore necessary to take measures to suppress the noise. So-called beam forming methods offer efficient solutions to these problems. Here, several microphones, so-called microphone arrays, are used for the reception of the speech signal. As a result of the spatial arrangement of the individual microphones with reference to the sound source, as well as due to the filtering and combination of the individual microphone signals, a spatial directive effect is produced. Signals that are incident on the microphone array from the useful signal direction are transferred essentially without distortion, while signals from other directions can be strongly suppressed. Adaptive beam-formers here can be adapted to movable interference sources that change over time, for example, the start phase, flight phase, landing phase, etc., of a plane. One prerequisite for the operation of a beam-former is to localize the speaker in the space, for example,

several pilots in a cockpit, and, optionally, to follow their movements. To achieve additional high directive effects, the filters in the beam-former must in part generate large amplifications. However, as a result, the sensitivity of the microphone array is increased with respect to individual microphones ~~of the microphone array~~, which are affected by error. Particularly serious interfering effects can result from tolerances in the transmission properties of the individual microphones, such as the frequency range, directive effect, sensitivity, etc.--

Page 2, first paragraph, amend:

--Thus, array microphones are capable of the targeted ~~resection~~ rejection of sound sources and speakers, short of the useful signals, and they can suppress interference signals, such as ambient noise or the generation of echo. Thus, for example, WO 99/39497 shows one possibility for the acoustical suppression of echoes for natural-speech installations. By means of this invention, undesired echoes that occur with natural-speech installations are to be eliminated. Here, an acoustical signal, a so-called pseudo noise signal, is emitted by a loudspeaker in the direction of at least two microphones. Adaptive filters, preferably FIR (finite impulse response) filters, are used to reshape the pseudo noise signal of a PN generator, by means of

algorithms that use a set of filter coefficients. The response signals of the microphones are combined by addition of the inverted output signals of the corresponding adaptive filters. Using LMS (least mean square) algorithms, the output signals of the adding device, that[[,]] is the combined signal, is adjusted such that its energy is minimal. For this purpose, the filter coefficients are changed.--

Page 2, third paragraph, amend:

--Array microphones essentially consist of an arrangement of individual microphones, which are interconnected by signal technology. In the arrangement of the microphones, one can distinguish, in principle, microphones that are in a one-, two-, and three-dimensional arrangement. In the one-dimensional arrangement, the microphones are strong along a line, for example, a straight line or an arc of a circle. When using <u>omni directional</u> microphones ~~with a spherical directive characteristic~~, the direction of the individual microphones is not essential because they only function as pressure receivers and their effect in space is therefore undirected--

Page 3, first paragraph, amend:

--When gradient microphones are used, the orientation of the

individual microphones is crucial: The overall directive characteristic and thus the overall ~~bundling~~ sound power concentration of the array microphone is produced by the combination of the directive characteristics of the individual microphones, using the algorithm, which is described in further detail below, by means of which the microphone signals are processed together.--

Page 3, seond paragraph, amend:

--One distinguishes two types of one-dimensional array microphones: broadside array microphones and endfire array microphones. They differ in the preferred direction of incidence of the sound with respect to the arrangement of the microphones: For endfire array microphones, the preferred direction of incidence of sound is in the longitudinal direction of the array microphone ~~microphones~~, that is, directions of incidences of sound with 0 **00.** For broadside array microphones, the preferred direction of incidence of sound is 0 = 900. The mutual ~~intervals~~ distances between the microphones can be constant or can differ from each other. In the second case, for different frequency ranges, different groups of microphones for the beam-forming are used, as described in M. Brandstein, D. Wards (Editors), Microphone Arrays, Springer Verlag, 2001.--

Page 3, third paragraph, amend:

--The connection, by signal technology, of the individual microphones can be analog or digital. Below, the digital implementation will be considered. The individual microsignals are digitized using A/D converters (analog/digital converters) and they are led to a signal processing unit. The signal processing unit uses an appropriate algorithm (key word "beam-forming") on the microphone signals. With the use of this algorithm, the ~~bundling degree~~ sound power concentration of the microphone is increased and lateral sound sources are suppressed. A good review of array microphones can also be found in M. Brandstein, D. Wards (Editors), Microphone Arrays, Springer Verlag, 2001 and in the literature cited therein.--

Paragraph bridging pages 3 aqnd 4, amend:

--Sets of filter coefficients are a component of the algorithm, and they are characteristic for the arrangement, the type, sensitivity, and characteristics of the microphones used, as well as the acoustical environment and the locations of the sound sources. Different properties of the different microphones, as produced, for example, by finishing dispersions, aging effects, etc., can be taken into account in these sets of filter coefficients. A frequently used ~~film~~s filter structure is

described in the literature under "Filter and Sum Beam-former" (see, for example, M. Brandstein, D. Wards (Editors), Microphone Arrays, Springer Verlag, 2001, page 159). Here, the individual microphone signals are filtered, after the analog/digital conversion, with appropriate FIR filters (finite impulse response filters) and then added. Fig. 1, which is representative of the state of the art, shows an embodiment example with 4 microphones.--

Page 4, first full paragraph, amend:

--Fig. 1 shows a simple array microphone ~~array~~ with identical distances d between the individual microphones. The incident angle of sound, 0, is expressed with reference to the longitudinal axis of the array microphone ~~array~~. The incident sound wave arrives ~~with~~ ~~after~~ different ~~travel times~~ time delays at the individual microphones of the array. The ~~travel time~~ time delay differences correspond to the path differences d*cos(0). The FIR filters 8 FIR_1 to FIR_4 shown in Fig. 1 contain filter coefficient sets that correspond to frequency-dependent differences in amplitude and phase. After the filtering, the signals are added (filter and sum beam-former). Due to the mentioned differences in amplitude and phase, the sound waves arriving at a certain direction of incidence are amplified by constructive ~~overlay~~ superposition, and sound waves coming ~~out of~~

from the other sound incidence ~~direction~~ directions are weakened by destructive overlaying. As the simplest special case, one can imagine the FIR filters 8 FIR_1 to FIR_4 to be so-called all-pass filters, all presenting the same frequency-independent delay. In this case, sound waves having an angle of incidence 0 = **900** are amplified, and sound waves from other directions of incidence are weakened, that is, the setup is that of a so-called broadside array.--

Page 4, third full paragraph, amend:

--The verification of individual microphones in the array occurs in such a manner that the current ~~uptake~~ consumption of the individual microphones is checked during the installation or during servicing. The value of the current ~~uptake~~ consumption is checked to determine whether it is between two predetermined limit values. In this manner, one can establish whether the individual microphone in principle is capable of operating. Nothing more happens.--

Page 5, second full paragraph, amend:

--One of these problems concerns the failure of an individual microphone. This can strongly decrease the ~~bundling degree~~ directivity index of the entire microphone and change the

directive characteristic in an undesired manner. The user observes a worsening of the function controlled by the array microphone, without being able to locate the precise cause, that is, the voice recognition suddenly works only poorly, and the speaker is poorly understood when telephoning.--

Page 5, third full paragraph, amend:

--In general, the poor performance results can have different causes, which do not have to be connected with the array microphone. For example, the GSM transmission line used during the telephoning can be defective. To allow a diagnosis of errors, it is therefore essential to know whether the array microphone is at least fully functional as a partial system. According to the state of the art, the current ~~uptake~~ consumption of the microphone can only be observed in the laboratory or during a service procedure.

Page 6, first full paragraph, amend:

--US 2002/0146136 Al discloses a method for the calibration of an acoustic converter, which is not part of an array microphone, in particular for mobile telephones. This calibration makes it possible for an electronic unit to deliver the desired amplitude and frequency responses, independently of the operative

differences that can occur between microphone and loudspeaker components. Here, a signal of a pseudo noise generator is applied through a filter to an external loudspeaker. The response signal of the microphone, in a DSP (digital signal processor), is filtered or converted using filter coefficients that reflect the inverse channel pulse response h of the arrangement; after filtering, it is compared with a "desired" signal obtained directly from the pseudo noise generator. The difference between the two signals, the so-called error signal, serves the function of changing the filter ~~coefficient~~ coefficients of the DSP. The filter is an adaptive type, that is, the filter coefficients are iteratively determined. They converge to a limit value, which results in the smallest possible error signal.--

Page 7, first paragraph, amend:

--An array microphone, which in its totality cannot be simply treated as the sum of its individual microphones, requires an entirely different testing from that of a single converter. Thus, during the installation of an array microphone, for example, in a vehicle cabin, the acoustic conditions are completely different compared to the test laboratory during development. Reflections, scattering, and interference due to multiple sound paths influence array microphones in a completely different manner than an individual microphone. In particular,

the directive characteristic and the ~~bundling degree~~ sound power concentration of the array microphone can dramatically change to the detriment of the user. Factors such as dust deposition on the membrane, changes in the polarization voltage, and similar factors, in the case of individual microphones, merely produce a slightly softer or duller output signal. In contrast, in array microphones, the same factors cause a change in the overall microphone characteristic, and they may even make the microphone unusable. The false polarity of an individual microphone, as a component of an array microphone, represents the worst case, where signals from the useful signal direction are largely suppressed.--

Page 9, first full paragraph, amend:

--Fig. 2 shows an embodiment example of an array microphone according to the invention, consisting of 4 microphones 1-4. The distances of the individual microphones 1-4 are the same in this embodiment example. The loudspeaker 5 is arranged in such a manner that its sound is acquired ~~it acquires sound~~ from all individual microphones 1-4, that is, a signal emitted by the loudspeaker 5 is received by all individual microphones. In variants, it is also possible to provide more than one loudspeaker, where it is not necessary that an individual microphone can receive the signals of all loudspeakers. It is

only important that all individual microphones can receive a signal from a loudspeaker. The individual microphones 1-4 can be designed either as pressure receivers or gradient receivers. Naturally, the invention is not limited to 4 individual microphones.--

Page 9, second full paragraph, amend:

--Fig. 3 shows an additional embodiment of the invention. In principle, the example has the same structure as in Fig. 2, but all the acoustic converters are accommodated in a common housing 6. In this housing, it is also possible to accommodate electronic components, AID and D/A converters 9, 10, digital filter 8, and signal processors 11. Only the openings, for speaking, of the microphones 1-4 are ~~shown~~ visible.

Paragraph bridging pages 9 and 10, amend:

--A calibration loudspeaker 5 – preferably a small loudspeaker based on the dynamic principle – is mounted in, on, or in the proximity of the array microphone, where the calibration loudspeaker has an acoustic connection to the individual microphones 1-4 of the array, in the sense that the loudspeaker's signal can be received by each of the individual microphones 1-4. For the case wherein only a single calibration

loudspeaker 5 is used, an appropriate place for its positioning is in the middle of the microphone arrangement, or in the plane of symmetry of the microphone arrangement, where the sum of all the calibration loudspeaker-individual microphone paths is at a minimum. However, other loudspeaker positions are also conceivable, for example, at the edge of the array or at some distance therefrom, as in the represented embodiment examples. The calibration ~~loudspeakers~~ loudspeaker is connected to an amplifier.--

Page 10, second full paragraph, amend:

--The purpose of the self-test of an array microphone according to the invention in particular involves the verification of one or more of the parameters of the individual microphones 1-4 listed below:

- The individual microphone is switched on,
- tue individual microphone has the correct polarity,
- the individual microphone has the desired sensitivity,
- the individual microphone presents the desired frequency ~~course~~ response of the sensitivity,
- the individual microphone does not present excessive distortion, and
- the ~~directed effect~~ directional behaviour of the individual microphone.--

Paragraph bridging pages 10 and 11, amend:

--Moreover, a self-test allows the determination of whether the individual microphones are in fact connected with the filters intended for them or whether connection errors occurred during the manufacturing process. For the purpose of verifying the individual microphone parameters, as listed above, the digital filters are programmed such that they represent an all-pass filter. The individual microphones can then reach the evaluation unit of the signal processor 11, in an "unbiased", that is,in the original, state. As a result of the relative position of the individual microphones with respect to each other, it is also possible for differences in ~~travel~~ time delay to be recorded.--

Page 13, first paragraph, amend:

--The evaluation of the ~~microphon~~ microphone signals can be carried out in different manners. As suitable measurement signals, one can use sinusoidal signals, stochastic noise signals, or periodic noise signals, such as maximum ~~cyclical noises~~ length sequences. Several methods are described below as examples:--

Page 13, third paragragpah, amend:

--Method 2) The loudspeaker sends out a periodic noise signal, for example, maximum ~~sequence noises~~ length sequences. By averaging the signal responses of the individual microphones, the signal/noise ratio is improved. From the averaged microphone signal responses, one can calculate the impulse responses of a given loudspeaker-microphone system using the so-called Fourier transformation (DFT). This method is analogous to the one found in the literature, for example, in Vorländer, M.: Anwendungen der Maximalfolgentechnik in der Akustik. Fortschritte der Akustik [Uses of the maximum sequence technique in acoustics. Progress in acoustics] – DAGA 94, pp. 83-102, for measuring loudspeakers and microphones. The impulse responses of the loudspeaker-microphone so measured are verified to determine whether their maximum is located within predetermined ~~travel times~~ values of time delay. The measured amplitude transfer functions are checked to determine whether they are within predeterniined tolerance ranges. These amplitude transfer functions are a measure of the microphone's sensitivity. By comparing with a reference measurement, it is possible to determine the change in microphone sensitivity caused, for example, by aging or environmental influences.--

Page 14, first paragraph, amend:

•- One embodiment variant of the acoustical self-

calibration consists of sending out a measurement signal that is inaudible to persons in the vicinity, for example, to the occupants of passenger cars. The measurement signal here is sent out in an audio range with a low level. By averaging the recorded microphone signals over time, measurements can be carried out even at signal/noise ratios <0 dB, as is the case in room acoustics measurements, for example, in fully occupied concert halls, during the performance. It is only after averaging the signal responses that the correlated signal portions are amplified and the uncorrelated background noise is eliminated.

- An additional embodiment variant consists of using several calibration loudspeakers; in this manner, the above-mentioned microphone parameters can be measured with greater precision and additional information on the ~~directive effect~~ directional behaviour of the microphones can be obtained
- Another embodiment variant of the acoustical self-calibration consists of carrying out the checking of the array in the ultrasound range, that is, using a frequency range that is inaudible to the user. For this purpose, the acoustical converters used must present, in a partial frequency range above 20 kJJz, sufficiently high transmission factors.--

Page 15, second full paragraph, amend:

--In this method according to the invention, the array microphones are automatically calibrated; the array niicrophones consists of several individual microphones 1-4, which are connected with a signal processor 11, which includes, for each individual microphone, at least one digital filter, where the signal processor 11 increases the ~~bundling degree~~ directivity index of the array microphone and suppresses lateral sound sources, by means of an appropriate algorithm applied to the individual microphone signals. In the process, filter coefficient sets, which are components of the algorithm, are applied to the digital filters, with the filter coefficient sets being characteristic for the arrangement, type, sensitivity, and characteristics of the individual microphones used, the acoustical environment, and the location of the sound sources. The signal processor 11 then proceeds to change the value of individual filter coefficients or of all the filter coefficients set, as a function of the deviation of the response signals from the model signals. The test can be repeated until the response signals are in the range of the model signals.--

Paragraph bridging pages 15 and 16, amend:

--It is possible for a person skilled in the art of electro-

acoustics to carry out this adaptation without problems if he/she is aware of the invention. It is preferred to carry out the selftest, new calculation and implementation at regular time intervals. This also allows an improvement of the microphone ~~bundling~~ sound power concentration, because it can be used to react to changing environmental conditions such as to the opening or closing of windows, to persons entering or leaving a vehicle, to changes in the microphone properties due to changes in the environmental parameters such as air temperature, air pressure or air humidity, direct exposure of a part of the array microphone to the sun, with the resulting differences in the heating of the individual microphones, etc.--

Page 16, second full paragraph, amend:

--If the loudspeakers are arranged clearly outside of the plane of symmetry of a linear array, as shown, for example, in Fig. 2 and Fig. 3, one has the possibility of carrying out the signal evaluation as described below. In the ideal case, the loudspeaker is mounted on the longitudinal axis of the microphone array outside of the microphone array itself. This method represents only an example of an evaluation; other arrangements are conceivable for a person skilled in the art who is aware of the invention. An all-pass filter with a ~~travel~~ time delay equal 0 ms is programmed into each filter of the individual microphone-

filter pairs. A periodic noise signal, for example, a Schröder noise with 8192 ~~scanning values~~ samples and a ~~scanning~~ sampling frequency of 44.1 kHz is applied to the loudspeaker. This corresponds to a period duration of 185.8 ins. The algorithm for generating Schröder noise is described, for example, in M. R. Schröder: Synthesis of Low-Peak-Factor Signals and Binary Sequences with Low Autocorrelation, IEEE transactions on information theory, pp. *85-89,* Vol. 16, January, 1970. The chosen period duration must be ~~louder~~ longer than or equal to the reverberation time RT_{60} of the measurement surrounding, for example, the cabin of a passenger car. This measurement signal is repeated, for example, 20 times, and acquired through the individual microphones and the associated filters. Here, the linearly measured sound pressure level ~~measured~~ at a 10-cm separation from the front edge of the loudspeaker is approximately 0.1 Pa.--

Paragraph bridging pages 16 and 17, amend:

--The following evaluation is then carried out for each microphone-filter pair: The signal is averaged, excluding the first period, synchronously to the input signal. The purpose of this averaging is to increase the signal/noise ratio, and thus to increase the precision of the measurement. Environmental noise, such as noise components of the microphone, the loudspeaker, and

the participating amplifiers, is suppressed by the averaging. The first period has to be excluded, because the first period contains a time section with uncorrelated signals due to the ~~ground noise~~ <u>inherent</u> delay that always exists.--

Page 17, first full paragraph, amend:

--The averaged signal response is subjected to ~~inverse~~ discrete Fourier transformation (IDFT) and the spectrum so obtained is divided by the IDFT of the excitation signal. The result then is the transfer function of the entire electroacoustic ~~four pole~~ <u>two-part system</u> loudspeaker-microphone-filter.--

Page 17, second full paragraph, amend:

--The amount of the transfer function must, in the case of a properly operating individual microphone, with a properly operating filter, ~~must~~ be within <u>a</u> predetermined tolerance ~~ranges~~ <u>range</u>.--

Page 17, fifth full paragraph, amend:

--In addition, it is possible to evaluate the ~~travel times~~ <u>time delays</u>. For this purpose, one transforms the transfer

function by inverse discrete Fourier transformation ~~(DFT)~~ (IDTF) into the time domain, and thus one obtains the impulse response of the entire electro-acoustical ~~four pole~~ two-part system loudspeaker-microphone-filter.--

Page 17, last paragraph, amend:

--From the impulse responses of the individual microphone-filter pairs, the corresponding ~~travel~~ time delays can easily be ascertained by determining the absolute maximum of the impulse responses. The ~~travel times~~ time delays of the individual microphone-filter pairs now must assume certain precalculated values as a function of the loudspeaker-microphone separation and as a function of the speed of sound in air. In particular, this makes it possible to determine whether individual microphones have been switched or whether the sequence of the microphones has been reversed by mistake.--